Full Featured IP PBX System for the Small Business and Home Office

DT-101

The IP-PBX the rich feature set of high-end PBX telephone systems with the convenience and cost advantages of Voice over IP. It has common voice system features such as an auto-attendant, shared line appearances, three way call conferencing, intercom, music on hold,call-forwarding and much more.

IT opens up access to the benefits of VoIP, including low cost long distance service, telephone number portability, and one network for both voice and data.It is so easy to configure that a fully working system can be set up in minutes. New telephones are automatically detected and registered when they are connected to the IP-PBX.The IP-PBX has an integrated web server that allow features to be configured using a web browser. The web server has multiple levels of password protected access to user and service level features. Service level settings may be locked by the Internet Telephone Service Provider to ensure they are not inadvertently corrupted. The Internet Telephone Service Provider also can remotely update the software and settings through a secure encrypted connection.

With its integrated router, it can be either connected directly to the internet connection or to another router on your network. IP-PBX has separate WAN and LAN Ethernet ports. The WAN connection can be connect through DHCP or a fixed IP address. The LAN port can assign IP addresses to IP telephones and computers using NAT and DHCP and will work with any SIP compatible IP telephone.

Powerful configuration capabilities enable the ip pbx to support a greater set of advanced features with these telephones, such as shared line appearances, hunt groups, call transfer, call parking lot, and group paging. With its two FXS ports, IP-PBX can support traditional analog devices such as telephones, answering machines, FAX machines, and media adapters.The IP-PBX initially supports four (4) SIP compatible IP Phones and is upgradeable to sixteen(16) IP Phones total per IP-PBX with an easy to install license key upgrade.

Features

• SIP Application Server, Proxy, Registrar and Location Server (RFC3261)
• Multiple Service Provider Lines / SIP Account Support (4)
• Shared Line Appearance (SLA)
• Automated Attendant (AA)
• Configurable AA Answer Delay
• Interactive Voice Response (IVR)
• Recordable IVR Prompts
• Automatic Call Distribution (ACD)
• Configurable Call Routing
- Least Cost Routing
- Multiple DID Numbers Per VoIP Line
- Call Routing to Multiple Extensions or Targeted User
- Call Hunting - Sequential, Round Robin, Random
• Phone Configuration and Management Server
- Discovery and Configuration of IP Phones
- Assignment of Extension
- Assignment of Dial plan
- Proxy Logging of SIP Messages
- Phone Firmware Upgrade Management
• Corporate Directory with Automatic Update
• Configuration and Maintenance via Web Interface (Local or Remote
- Status Display of All Connections
• Remote Configuration via
- HTTPS with XML Formatted Files
- HTTP or TFTP with 256-Bit Encrypted Binary Files
• Call Park - User Definable Parking Space Number
• Call Unpark
• Call Transfer
• Call Forward
• Group Paging
• Intercom
• Directed Call Pick Up
• Group Call Pick Up
• Music / Information via Streaming Audio Server (SAS) for Calls:
- On Hold
- Parked in the Parking Lot
- Being Transferred
• Simultaneous Ringing (Find Me Service)
• Do Not Disturb
• Voice Mail Integration - Service Provider Based
- Voice Mail Notification via SUBSCRIBE / NOTIFY
- Forward Call Directly to Voice mail
• Integrated Media Proxy or Direct RTP Routing to ITSP
• Differentiated Services (DiffServ) / Type of Service (TOS) Support
• Two FXS Ports for Phones, Fax machines, Media Adapters
• Voice encoding according to G.711 (64kbit/s)
• Fax Support using G.711 Pass-Through or T.38
• Echo Cancellation (G.165)

Security

Password Protected System Reset to Factory Default
• Password Protected Admin and User Access Authority
• HTTPS with Factory Installed Client Certificate
• HTTP Digest - Encrypted Authentication via MD5 (RFC 1321)
• Up to 256-bit AES Encryption

LEDs

• Power, Internet, Phone 1, Phone 2

Package Contents

• 1 – IP-PBX
• 1 - 5 Volt Power Adapter
• 1 - RJ45 Ethernet Cable
• 1 - Quick Installation

Environmental

Dimensions
Unit Weight
Power
Operating Temp.
Storage Temp.
Operating Humidity
Storage Humidity
3.98 x 3.98 x 1.1 in (101 x 101 x 28 mm) W x H x D
5.3 oz (0.15 kg)
Switching Type (100-240v) Automatic, DC Input Voltage: +5 VDC at 2.0 A Max.,
Power Consumption: 5 Watts, Power Adapter: 100-240v - 50-60Hz (26-34VA) AC Input,
1.8m cord
32؛۱۱۳~؛F (0؛۴۵~؛C)
-13؛۱۸۵~؛F (-25؛۸۵~؛C)
10~90% Non-condensing
10~90% Non-Condensing

Specifications

Data Networking
MAC Address (IEEE 802.3)
IPv4 - Internet Protocol v4 (RFC 791) upgradeable to v6 (RFC 1883)
ARP - Address Resolution Protocol
DNS - A Record (RFC 1706), SRV Record (RFC 2782)
DHCP Client - Dynamic Host Configuration Protocol (RFC 2131)
DHCP Server - Dynamic Host Configuration Protocol (RFC 2131)
PPoE Client - Point to Point Protocol over Ethernet (RFC 2516)
ICMP - Internet Control Message Protocol (RFC792)
TCP - Transmission Control Protocol (RFC793)
UDP - User Datagram Protocol (RFC768)
RTP - Real Time Protocol (RFC 1889) (RFC 1890)
RTCP - Real Time Control Protocol (RFC 1889)
DiffServ (RFC 2475), Type of Service - TOS (RFC 791/1349)
VLAN Tagging - 802.1p/q
SNTP - Simple Network Time Protocol (RFC 2030)
Upload Data Rate Limiting - Static and Automatic
QoS - Voice Packet Prioritization over Other Packet Types
Router or Bridge Mode of Operation
MAC Address Cloning
Port Forwarding
Voice Gateway

SIPv2 - Session Initiation Protocol Version 2 (RFC 3261, 3262, 3263, 3264)
SIP Proxy Redundancy - Dynamic via DNS SRV, A Records
Re-registration with Primary SIP Proxy Server
SIP Support in Network Address Translation Networks - NAT (incl. STUN)
Secure (Encrypted) Calling via Pre-Standard Implementation of Secure RTP
Codec Name Assignment
Voice Algorithms:
- G.711 (A-law and μ-law)
- G.726 (16/24/32/40 kbps)
- G.729 A
- G.723.1 (6.3 kbps, 5.3 kbps)
Dynamic Payload Support
Adjustable Audio Frames Per Packet
DTMF: In-band & Out-of-Band (RFC 2833) (SIP INFO)
Flexible Dial Plan Support with Inter-Digit Timers
IP Address / URI Dialing Support
Call Progress Tone Generation
Jitter Buffer - Adaptive
Frame Loss Concealment
VAD - Voice Activity Detection w/ Silence Suppression
Attenuation / Gain Adjustments
MWI - Message Waiting Indicator Tones
VMWI - Via NOTIFY, SUBSCRIBE
Caller ID Support (Name & Number)
Provisioning,Administration &Maintenance:
Web Browser Administration & Configuration via Integral Web Server
Telephone Key Pad Configuration of Select Networking Parameters via IVR
Automated Provisioning & Upgrade via HTTPS, HTTP, TFTP
Asynchronous Notification of Upgrade Availability via NOTIFY
Non-intrusive, In-Service Upgrades
Report Generation & Event Logging
Stats in BYE Message
Syslog & Debug Server Records - Per Line Configurable
Physical Interfaces
2 10/100BaseT RJ-45 Ethernet Port (IEEE 802.3) -- 1 WAN, 1 LAN
2 RJ-11 FXS Phone Ports - For Analog Circuit Telephone Device (Tip/Ring)
Subscriber Line ,Interface Circuit,(SLIC):

Ring Voltage: 40-55 VRMS Configurable
Ring Frequency: 10 Hz - 40 Hz
Ring Waveform: Trapezoidal and Sinusoidal
Maximum Ringer Load: 3 REN
On-hook/off-hook Characteristics: On-hook voltage (tip/ring): -50 V NOMINAL, Off-hook current: 25 mA min, Terminating Impedance: 8 Configurable Settings including North America 600 ohms, European CTR21Switching Type (100-240v) Automatic